Avaya Solution & Interoperability Test Lab

Configuring Avaya Communication Manager, Cisco 2811 Gatekeeper and Cisco CallManager to Control VoIP Calls between Avaya and Cisco Telephones - Issue 1.0

Abstract

These Application Notes describe a configuration, including, Avaya Communication Manager, a Cisco 2811 Gatekeeper, and Cisco CallManager that was verified for a customer proof of concept request. In this configuration, Avaya Communication Manager and Cisco CallManager are registered with the Cisco 2811 Gatekeeper, which provides call control between Avaya Communication Manager and Cisco CallManager.

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1. Introduction

An H.323 gatekeeper can provide address translation and network access control services for H.323 terminals and gateways. These Application Notes present the configuration steps to allow Avaya Communication Manager to communicate with Cisco CallManager via a Cisco 2811 Gatekeeper. These Application Notes will cover the necessary configuration steps for Avaya Communication Manager, a Cisco 2811 Gatekeeper and Cisco CallManager.

Figure 1 depicts the network configuration utilized for these Application Notes.
In Location A in **Figure 1**, the Avaya G350 Media Gateway with S8300 Server provides both H.323 Gateway (GW) and Gatekeeper (GK) functionality. In Location B, the Cisco 2811 access router is configured as an H.323 Gatekeeper, and the Cisco CallManager is registered with it. The Cisco 2811 GK provides E.164 to IP address resolution and call control for bandwidth between the Location A and Location B.

The are two zones defined on this Gatekeeper, a local zone for Location B and a remote zone for Location A. The Gatekeeper stays in Local zone. The following shows the basic flow for a call from Location A to Location B:

- Avaya Communication Manager sends a Location Request (LRQ) message to the Cisco 2811 Gatekeeper.
- The Cisco 2811 Gatekeeper checks its dial plan and bandwidth requirement
- If the call meets the bandwidth requirement:
    - The Cisco 2811 GK responds with location confirm (LCF) to Avaya Communication Manager.
    - Avaya Communication Manager will set up the call to the Cisco CallManager through H.225 (Q.931) signaling.
- If the bandwidth requested exceeds the bandwidth requirement, the Cisco 2811 Gatekeeper sends call Location Rejection (LRJ) to Avaya Communication Manager to deny the call.

Similarly, this call flow can be applied for call origination in Location B to Location A.

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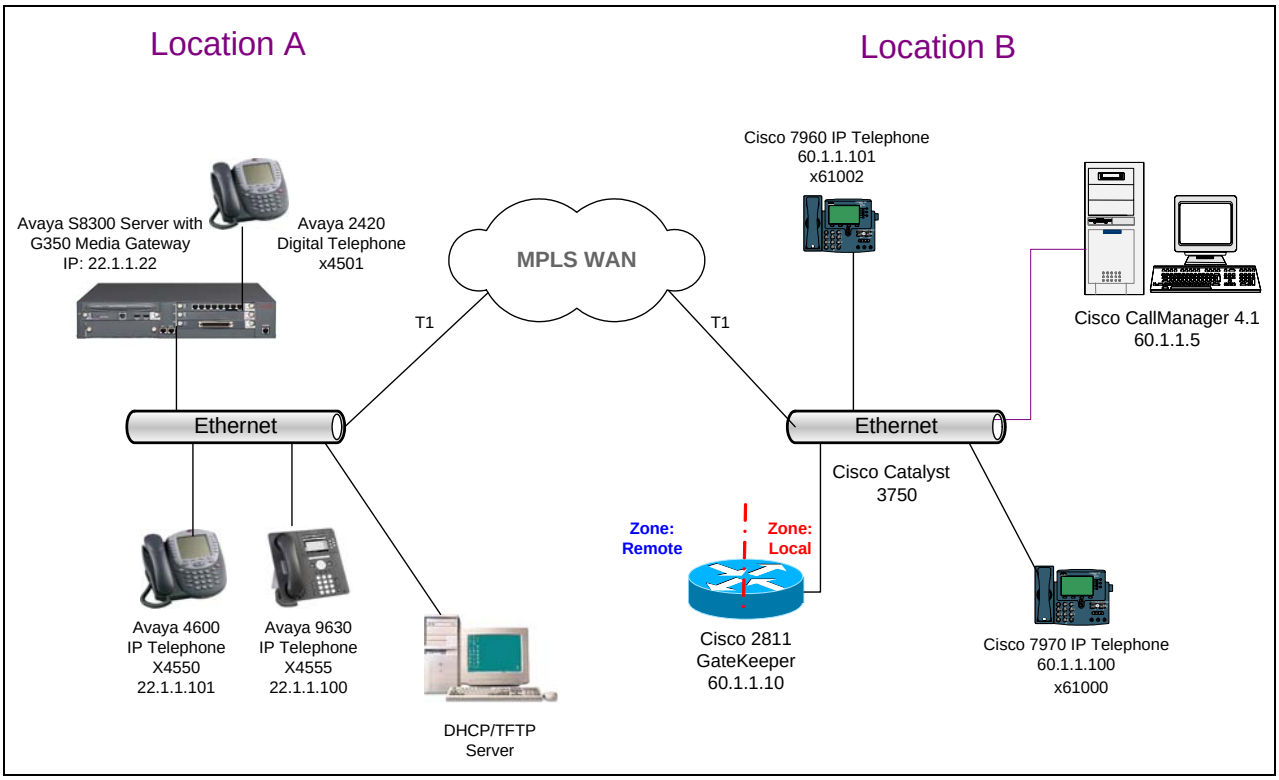

Figure 1: Network Diagram for these Application Notes

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Network Component	Software Version
Avaya Communication Manager S8300 Server with G350 Media Gateway	3.1.2 (634)
Avaya G350 Media Gateway	26.31
Avaya 9600 IP Telephone (H.323)	1.2.1
Avaya 4620 IP Telephone (H.323)	2.7
Cisco 2811 Gatekeeper	12.4(2)T5 (IP Voice)
Cisco CallManager	4.1(2)
Cisco Catalyst 3750 Switch	8.2.1
Cisco 7970/7960 IP Telephones	7.2

Table 1 - Network Component Software Versions

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3. Avaya Communication Manager Configuration

3.1. Configuring Node Names on the Avaya Communication Manager

This section presents configuration steps for node name to IP address mapping on Avaya Communication Manager. It is assumed that appropriate license and authentication files have been installed on the S8300 Server. It is also assumed that the network IP address on the S8300 Server has been configured to 22.1.1.22/24 and the IP address on the G350 GW has been configured to 22.1.1.23/24.

Avaya Communication Manager SAT screens can be accessed using "telnet 192.11.13.6 5023" from a computer connected to the services port, or "telnet 22.1.1.22 5023" through the network.

Use the command **change node-names ip** to display the mapping of node names to IP addresses. The node name **Call-Manager** is mapped to Cisco CallManager's IP address **60.1.1.5** and the node name **Cisco-GK** is mapped to the Cisco 2811 router's IP address **60.1.1.10**. The **procr** is mapped to the Avaya S8300 Server's IP Address **22.1.1.22**.

```
change node-names ip                                    Page   1 of   1
                                IP NODE NAMES
    Name                 IP Address          Name              IP Address
Call-Manager          60 .1  .1  .5                             .   .   .
Cisco-GK              60 .1  .1  .10                            .   .   .
procr                 22 .1  .1  .22                            .   .   .
```

3.2. Configuring IP Network Region

In this sample configuration, the IP network region 1 is configured for Location A and the IP network region 2 is configured for Location B.

Use the command **change ip-network-region 1** to specify the IP codec set to be used for network region 1 and the IP codec set used between network region 1 and region 2. Page 1 of 19 shows that codec set **1** is used for network region **1**. The fields under **DIFFSERV/TOS PARAMETERS** and **802.1P/Q PARAMETERS** are used for QoS. The Avaya IP telephone will receive these QoS parameters upon registration.

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```
change ip-network-region 1                                    Page   1 of  19
                              IP NETWORK REGION
  Region: 1
Location:                   Home Domain:
    Name:
                                 Intra-region IP-IP Direct Audio: no
AUDIO PARAMETERS                 Inter-region IP-IP Direct Audio: no
   Codec Set: 1                            IP Audio Hairpinning? n
UDP Port Min: 16384
UDP Port Max: 32767                        RTCP Reporting Enabled? y
                                RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS          Use Default Server Parameters? y
 Call Control PHB Value: 34
        Audio PHB Value: 46
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 3
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
    Keep-Alive Count: 5
```

Page 3 of 19 shows that calls between network region **1** and **2** are allowed, up to the **256:Kbps**. Calls above the specified limit will not be permitted, and may be re-routed to a different path. The bandwidth usage is based on the packet size and codec selection. Avaya supports IP Call Admission Control via Bandwidth Limits (CAC-BL). Codec set **2** is used for calls between regions 1 and 2.

```
change ip-network-region 1                                    Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   Total           Video                            Dyn
 rgn rgn  set   WAN  WAN-BW-limits  WAN-BW-limits  Intervening-regions  CAC IGAR
 1   1    1
 1   2    2     y     256:Kbits        0:Kbits                             n
```


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Use the command **change ip-network-region 2** to specify the IP codec set to be used for network region 2 and the IP codec set used between network region 1 and region 2. Page 1 of 19 shows that codec set **2** is used for network region **2**.

```
change ip-network-region 2                                  Page   1 of  19
                                IP NETWORK REGION
  Region: 2
Location:                   Home Domain:
    Name:
                                 Intra-region IP-IP Direct Audio: no
AUDIO PARAMETERS                 Inter-region IP-IP Direct Audio: no
   Codec Set: 2                            IP Audio Hairpinning? n
UDP Port Min: 16384
UDP Port Max: 32767                        RTCP Reporting Enabled? y
                                  RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS            Use Default Server Parameters? y
 Call Control PHB Value: 34
        Audio PHB Value: 46
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 3
        Audio 802.1p Priority: 6      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
    Keep-Alive Count: 5
```

Page 3 of 19 shows that calls between network region **2** and **1** are allowed, up to the **256:Kbps**. Calls above the specified limit will not be permitted, and may be re-routed to a different path. The bandwidth usage is based on the packet size and codec selection. Avaya supports IP Call Admission Control via Bandwidth Limits (CAC-BL). Codec set **2** is used for calls between regions 2 and 1.

```
change ip-network-region 2                                    Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   Total           Video                            Dyn
 rgn rgn  set   WAN  WAN-BW-limits  WAN-BW-limits  Intervening-regions  CAC IGAR
 2   1    2     y     256:Kbits        0:Kbits                            n
 2   2    2
```


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Use the command **change ip-codec-set x** (where x is the codec-set number) to configure IP codec(s). The following screens show the codec configuration. IP-codec-set 1 uses **G.711MU** and ip-codec-set 2 uses **G.729**.

```
change ip-codec-set 1                                          Page   1 of   1
                         IP Codec Set
    Codec Set: 1

    Audio      Silence      Frames   Packet
    Codec      Suppression  Per Pkt  Size(ms)
 1: G.711MU         n         2        20
 2:
```

```
change ip-codec-set 2                                          Page   1 of   1
                         IP Codec Set
    Codec Set: 2

    Audio      Silence      Frames   Packet
    Codec      Suppression  Per Pkt  Size(ms)
 1: G.729       n         2        20
 2:
```

3.3. Configuring Signaling and Trunk Groups to Cisco 2811 Gatekeeper

In order to configure Avaya Communication Manager as an H.323 Gatekeeper to send LRQ to the Cisco 2811 GK, a signaling group for outbound traffic must be configured. Note that the IP address of the Avaya S8300 Server (procr) functions as an IP address for both H.323 GW and GK.

Use the command **add signaling-group** <group number> to create a signaling group. The Near-End Node Name and Far-end Node Name are configured to the **procr** and **Cisco-GK** respectively. To configure the Avaya S8300 Server as a H.323 GK, LRQ required must be set to **y**. Near-end Listen Port and Far-end Listen Port must be configured to **1719** (UDP port 1719 for H.225 RAS). Direct IP-IP Audio Connections and IP Audio Hairpinning must be set to **n** for successful media paths. Disabling Direct IP-IP and IP Audio hairpinning will establish the RTP stream between the G350 Media Gateway and the Cisco IP Telephones for inter location calls. To support out-of-band DTMF, DTMF over IP must be configured to **out-of-band**. To control Codec Set selection, the Far-end Network Region is configured to **2**. Note that this signaling group is configured to be associated with the trunk group 20.

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```
add signaling-group 20                                    Page   1 of   5
                              SIGNALING GROUP

 Group Number: 20             Group Type: h.323
                            Remote Office? n          Max number of NCA TSC: 20
                                      SBS? n           Max number of CA TSC: 0
                                 IP Video? n        Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 20
          Supplementary Service Protocol: a
                          T303 Timer(sec): 10

   Near-end Node Name: procr                 Far-end Node Name: Cisco-GK
 Near-end Listen Port: 1719                Far-end Listen Port: 1719
                                         Far-end Network Region: 2
         LRQ Required? y                  Calls Share IP Signaling Connection? n
         RRQ Required? n                        H245 Control Addr On FACility? n
     Media Encryption? n                       Bypass If IP Threshold Exceeded? n
                                                       H.235 Annex H Required? n
         DTMF over IP: out-of-band              Direct IP-IP Audio Connections? n
                                                          IP Audio Hairpinning? n
                                                  Interworking Message: PROGress
                                          DCP/Analog Bearer Capability: 3.1kHz
```

Use the command **add trunk–group <group number>** to add an IP trunk group. The following screen shows the configuration of the IP trunk group 20. As a standard IP trunk configuration, Group Type is configured to **isdn**, Carrier Medium to **H.323** and Service Type to **tie**.

```
add trunk-group 20                                        Page   1 of  21
                                TRUNK GROUP

Group Number: 20                  Group Type: isdn          CDR Reports: y
  Group Name: TO Ciso-GK                  COR: 1        TN: 1        TAC: 120
   Direction: two-way         Outgoing Display? n         Carrier Medium: H.323
 Dial Access? n                 Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                                     Member Assignment Method: manual
```


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Note that Codeset to Send Display must be configured to **0** in order for the Cisco GW to accept the Q.931 call setup message from the Avaya Media Gateway.

```
add trunk-group 20                                        Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 0     Codeset to Send National IEs: 6
                                        Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

                                                 QSIG Value-Added? n
                                               Digital Loss Group: 18
Incoming Calling Number - Delete:     Insert:                Format: unk-pvt
```

In order to support caller ID on the IP trunk (isdn type), configure Send Name and Send Calling Number to **y**. In the sample configuration, the Format is configured to **unk-pvt**.

```
add trunk-group 20                                        Page   3 of  21
TRUNK FEATURES
       ACA Assignment? n           Measured: none
                                Internal Alert? n        Maintenance Tests? y
                                Data Restriction? n     NCA-TSC Trunk Member:
                                     Send Name: y      Send Calling Number: y
            Used for DCS? n                           Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: unk-pvt
                                        UUI IE Treatment: service-provider

                                             Replace Restricted Numbers? n
                                            Replace Unavailable Numbers? n
                                                  Send Connected Number: n
                                              Hold/Unhold Notifications? n
             Send UUI IE? y                 Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y
```


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IP trunk member port values are initially configured to “ip” in the Port field. The system will then assign a value to the port as shown below. Note the IP trunk members are virtual ports. In the following screen, five members associated with signaling group 20 are configured. This means that the IP trunk group only supports five simultaneous VoIP calls. The sixth call will be blocked if there is no backup trunk available. If a PSTN trunk is configured as a backup trunk, the additional calls can be overflowed to the PSTN, or all calls can use the PSTN if the VoIP trunk is out of service.

```
display trunk-group 20                                    Page   5 of  21
                                TRUNK GROUP
                                     Administered Members (min/max):   1/5
GROUP MEMBER ASSIGNMENTS                 Total Administered Members:   5

       Port              Name        Night           Sig Grp
  1: T00086                                            20
  2: T00087                                            20
  3: T00088                                            20
  4: T00089                                            20
  5: T00090                                            20
  6:
```

When a call is placed from Avaya Communication Manager, the called number is routed (by appropriate routing, e.g., UDP, AAR/ARS, etc.) to the Cisco 2811 Gatekeeper via trunk group 20. The Cisco 2811 Gatekeeper will send the call setup message to the appropriate gateway, for these Application Notes, Cisco CallManager.

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Separate signaling and trunk groups are required for Avaya Communication Manager to receive calls. When Avaya Communication Manager receives a call setup message from a Cisco CallManager via the Cisco 2811 Gatekeeper, Avaya Communication Manager will try to match it to a signaling group based on the remote GW's IP address. The call will be denied if there is no match. To reduce the configuration required, the Far-end Node Name can be left unspecified, or an inbound signaling group can be configured for each GW from which Avaya Communication Manager should accept calls. The following screen shows the configuration for inbound signaling group 21. Note that Near-end Listen Port must be **1720** to accept incoming calls (TCP port 1720 for H.225 call setup messages). The LRQ Required is set to **n**. Like the configuration for signaling group 20, Direct IP-IP Audio Connections and IP Audio Hairpinning are set to **n**, DTMF over IP is configured to **out-of-band,** and the Far-end Network Region is set to **2**.

```
display signaling-group 21                                    Page   1 of   5
                               SIGNALING GROUP

 Group Number: 21             Group Type: h.323
                            Remote Office? n         Max number of NCA TSC: 0
                                      SBS? n          Max number of CA TSC: 0
                               IP Video? n       Trunk Group for NCA TSC: 20
      Trunk Group for Channel Selection: 21
         Supplementary Service Protocol: a
                        T303 Timer(sec): 10

   Near-end Node Name: procr                 Far-end Node Name:
 Near-end Listen Port: 1720                Far-end Listen Port:
                                        Far-end Network Region: 2
        LRQ Required? n                Calls Share IP Signaling Connection? n
        RRQ Required? n                      H245 Control Addr On FACility? n
    Media Encryption? n                     Bypass If IP Threshold Exceeded? n
                                                    H.235 Annex H Required? n
        DTMF over IP: out-of-band          Direct IP-IP Audio Connections? n
                                                     IP Audio Hairpinning? n
                                              Interworking Message: PROGress
                                       DCP/Analog Bearer Capability: 3.1kHz
```

The following screen captures show the configuration of an IP trunk group 21 for incoming calls from Location B.

```
display trunk-group 21                                        Page   1 of  21
                               TRUNK GROUP

Group Number: 21                   Group Type: isdn          CDR Reports: y
  Group Name: OUTSIDE CALL               COR: 1        TN: 1        TAC: 121
   Direction: two-way        Outgoing Display? n        Carrier Medium: H.323
 Dial Access? n                Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                             Member Assignment Method: manual
```


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```
display trunk-group 21                                          Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 0     Codeset to Send National IEs: 6
                                        Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc

                                                  QSIG Value-Added? n
                                                Digital Loss Group: 18
Incoming Calling Number - Delete:     Insert:                 Format: unk-pvt

 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

```
display trunk-group 21                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n           Measured: none
                                Internal Alert? n        Maintenance Tests? y
                              Data Restriction? n     NCA-TSC Trunk Member:
                                     Send Name: y      Send Calling Number: y
            Used for DCS? n                           Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: unk-pvt
                                  UUI IE Treatment: service-provider

                                        Replace Restricted Numbers? n
                                       Replace Unavailable Numbers? n
                                             Send Connected Number: y
                                        Hold/Unhold Notifications? y
             Send UUI IE? y                Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y
```

```
display trunk-group 21                                          Page   5 of  21
                                TRUNK GROUP
                                     Administered Members (min/max):   1/5
GROUP MEMBER ASSIGNMENTS                 Total Administered Members:   5

       Port             Name        Night           Sig Grp
  1: T00091                                           21
  2: T00092                                           21
  3: T00093                                           21
  4: T00094                                           21
  5: T00095                                           21
  6:
```


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3.4. Configuring Dial Plan and Caller ID on Avaya Communication Manager

Uniform Dialing Plan (UDP) and Automated Alternate Routing (AAR) are used for the call routing. The configurations in this section are for demonstration only.

Use the command **display system-parameters customer-options** to verify that Uniform Dialing Plan is set to **y**.

```
display system-parameters customer-options                    Page   5 of  10
                               OPTIONAL FEATURES

                    Posted Messages? n                Tenant Partitioning? n
                    PNC Duplication? n         Terminal Trans. Init. (TTI)? n
                Port Network Support? y                Time of Day Routing? n
                                                     Uniform Dialing Plan? y
           Processor and System MSP? n       Usage Allocation Enhancements? y
                 Private Networking? y           TN2501 VAL Maximum Capacity?
…
```

Use the command **change dialplan analysis** to configure the dial plan. The dialed digits **61**xxx ("x" is any digit) is configured as **ext**. Note the digits 61xxx match the extensions created on the Cisco CallManager.

```
change dialplan analysis                                      Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                                Percent Full:  1

      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        1       3    dac
        2       4    ext
        61      5    ext
```


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Use the command **change aar analysis 6** ("6" is the first dialed digit) to configure the dialed digits **61**xxx to use **aar** and route pattern **20**.

```
change aar analysis 6                                    Page   1 of   2
                          AAR DIGIT ANALYSIS TABLE
                                                      Percent Full:    1

          Dialed          Total     Route    Call   Node  ANI
          String         Min  Max  Pattern   Type   Num   Reqd
    61                    5    5     20        aar          n
…
```

Use the command **change route-pattern 20** to set the trunk group 20 as a primary trunk group to carry the calls to 61xxx. Since there are no digits deleted or inserted, all 5 digits will be used as the called number. The Numbering Format is configured to **lev0-pvt** to support caller ID in the sample configuration.

```
change route-pattern 20                                  Page   1 of   3
                    Pattern Number: 20  Pattern Name: cisco-GK
                              SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                              Dgts                                  Intw
 1: 20   0                                                           n   user
 2:                                                                  n   user
 3:                                                                  n   user
 4:                                                                  n   user
 5:                                                                  n   user
 6:                                                                  n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                   Dgts Format
                                                           Subaddress
 1: y y y y y n  n           rest                                 lev0-pvt  none
 2: y y y y y n  n           rest                                           none
 3: y y y y y n  n           rest                                           none
```

In order to support caller ID for the private network, use the command **display system-parameters customer-options** to verify that Private Networking is set to **y**.

```
display system-parameters customer-options                  Page   5 of  10
                              OPTIONAL FEATURES

                    Posted Messages? n                  Tenant Partitioning? n
                    PNC Duplication? n           Terminal Trans. Init. (TTI)? y
               Port Network Support? y                    Time of Day Routing? n
                                                        Uniform Dialing Plan? y
            Processor and System MSP? n          Usage Allocation Enhancements? y
                  Private Networking? y            TN2501 VAL Maximum Capacity? n
…
```


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Use the command **change private-numbering** to change private format for the calling number. The configuration below will use the phone extension as the calling number.

```
change private-numbering                                        Page   1 of   1

                           NUMBERING - PRIVATE FORMAT

              Network Level: 0              PBX Identifier:
               Level 2 Code:                Deleted Digits: 0
               Level 1 Code:
```

The command **save translation** must be entered to save the administration performed.

4. Cisco 2811 Gatekeeper and Cisco CallManager Configuration

4.1. Configuring the Cisco 2811 Gatekeeper

The following are the annotated Cisco 2811 Gatekeeper configuration. A Gatekeeper will forward the LRQ from Avaya CM and Cisco CallManager when the LRQ forwarding is enabled on the GK. To enable a gatekeeper to forward LRQ messages, use the **lrq forward-queries** command in gatekeeper configuration mode. Note that **lrq forward-queries add-hop-count** must be configured on the Cisco 2811 GK, to allow the Cisco 2811 GK to forward the LRQ from Avaya Communication Manager.

There is one local zone named **GateKeeper** and one remote zone named **G350** configured on the Cisco 2811 GK. Zone prefix “61*” (where * will match any sequence of numbers following 61) is associated with the Cisco CallManager telephone extensions, and the Zone prefix “45*” is associated with the extensions configured on Avaya Communication Manager.

For bandwidth control, the maximum bandwidth allowed between the local and remote zones is configured as 256 kb. If the maximum bandwidth is fully utilized, the GK will reject any additional call attempts.

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voice call carrier capacity active !	
interface FastEthernet0/0 **ip address 60.1.1.10 255.255.255.0** duplex auto speed auto ! ! ip classless ip route 0.0.0.0 0.0.0.0 60.1.1.2	*Interface FastEthernet 0/0 configuration.*
gatekeeper **zone local GateKeeper ABC.com 60.1.1.10**	*Enter Gatekeeper configuration* *Configure local zone GateKeeper with IP address 60.1.1.10.*
zone remote G350 ABC.com 22.1.1.22 1719	*Configure remote zone G350 with IP address 22.1.1.22.*
zone prefix G350 45*	*Configure zone prefix for remote GK G350. Gatekeeper will route calls starting with digits 45 to G350 Media Gateway.*
zone prefix GateKeeper 61*	*Configure Cisco CallManager prefix to 61*. Gatekeeper will route calls starting with digits 61 to Cisco CallManager.*
gw-type-prefix 1#* default-technology gw ipaddr 60.1.1.5 1720	*Configure 1#* as default-technology for Cisco CallManager.*
lrq forward-queries add-hop-count	*Enable lrq forward*
bandwidth remote 256	*Configure the Max. bandwidth to remote zone 256kb.*
bandwidth check-destination	*Enable bandwidth checking before permit call.*
no shutdown	*Enable Gatekeeper*

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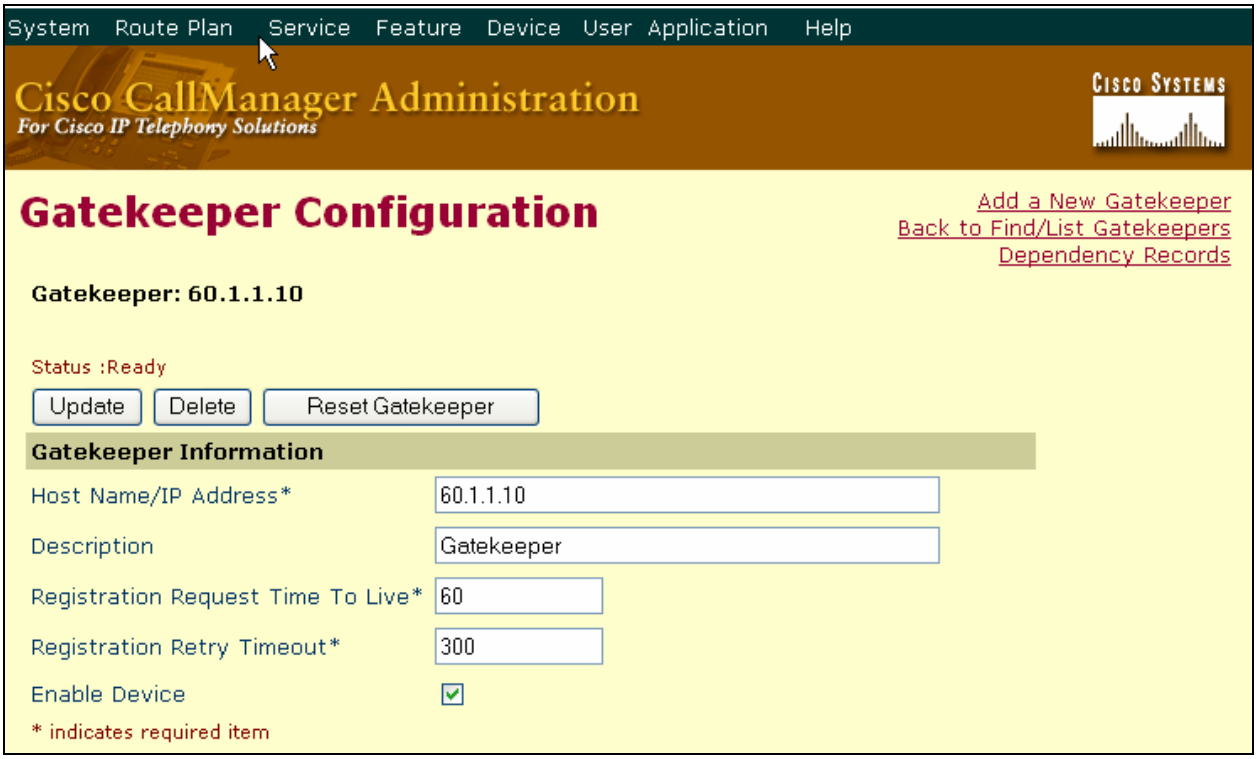
4.2. Configuring Cisco CallManager

To use the Gatekeeper to control calls from the Cisco CallManager, the Cisco 2811 router must be added on the Cisco CallManager as a Gatekeeper. Configurations are shown below.

4.2.1. Add Gatekeeper

Launch Cisco CallManager Administration Application and log in with proper user name and password. The display below shows the gatekeeper configuration. To add a gatekeeper, follow the following steps.

- Click **Device → Gatekeeper**
- Enter the Cisco 2811 Gatekeeper's IP address **60.1.1.10** as **Host Name/IP Address**
- Enter **Gatekeeper** for **Description** (Optional)
- Leave **Registration Request Time To Live** and **Registration Retry Timeout** as default
- Check box **Enable Device**
- Click **Update**
- Click **Reset Gatekeeper**



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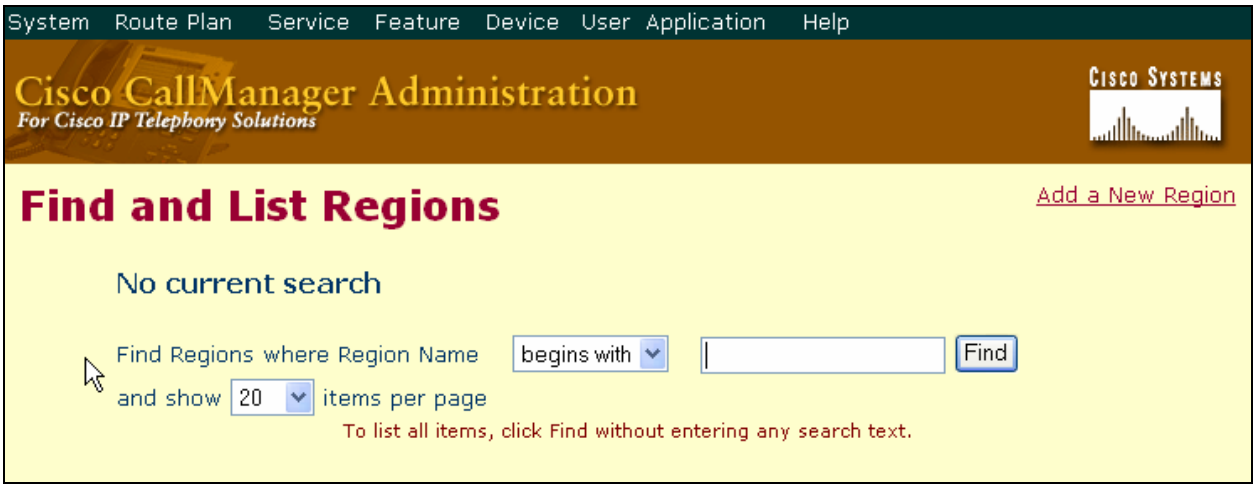
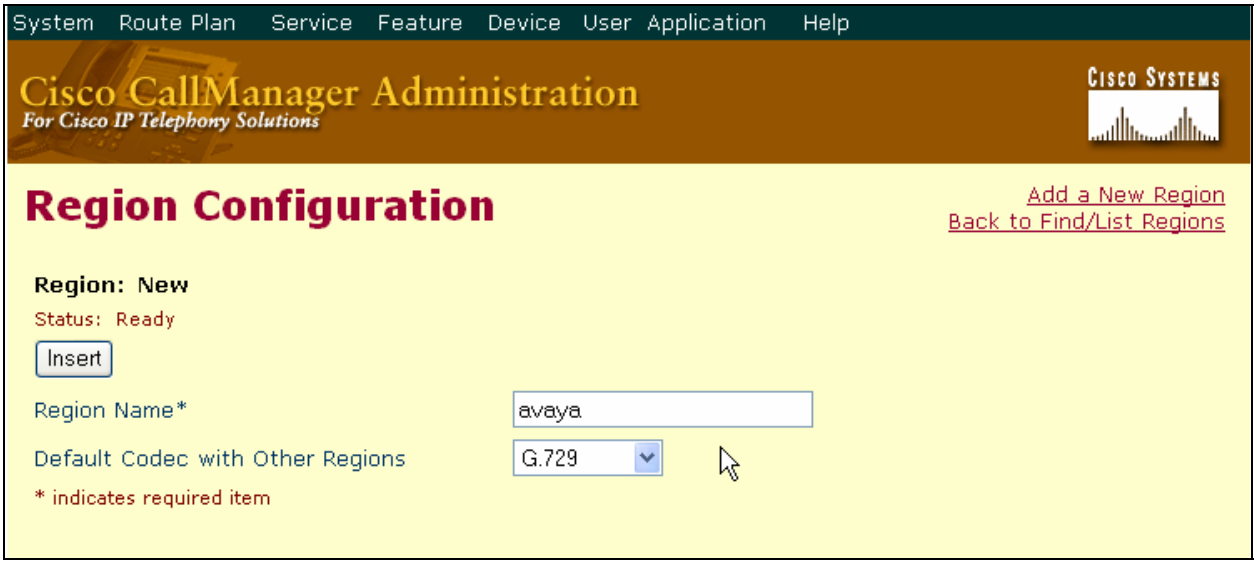
4.2.2. Add New Region

Regions are used for codec selection. In these Application Notes, two regions are used. The default region is used for Location B and a new region, named Avaya, is created for Location A. To save bandwidth on WAN link, codec G.729 is used between these two regions. Intra-region calls will use codec G.711 ulaw. The following steps show how to create a new region on Cisco CallManager.

- Click **System → Region**
- Click **Add a New Region** as shown below


System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Cisco Systems
Find and List Regions
Add a New Region
No current search
Find Regions where Region Name
begins with
Find
and show 20 items per page
To list all items, click Find without entering any search text.

- Enter **avaya** as **Region Name**
- Select **G.729** as **Default Codec with Other Regions.** This will force the Cisco CallManager to use codec G.729 for calls to other region.
- Click **Insert**



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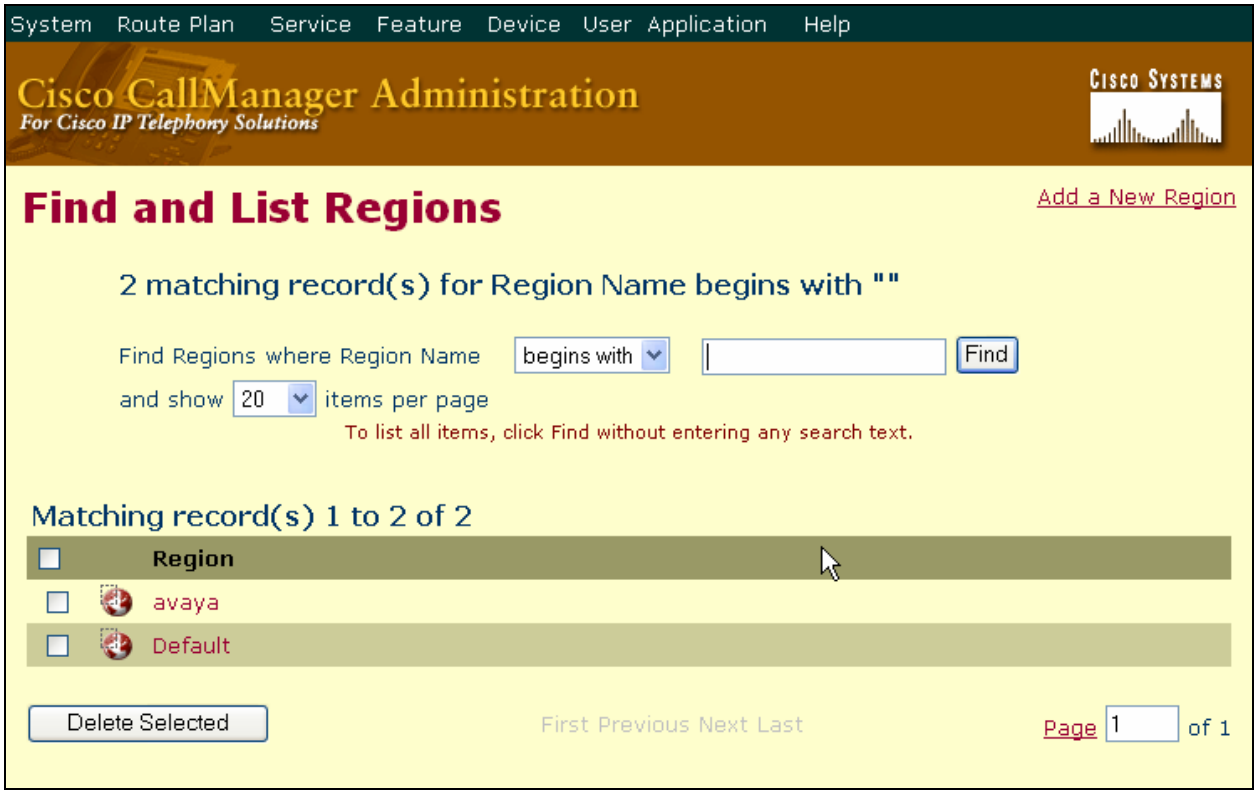
- Click **System → Region**
- Click **Find** and two regions are displayed
- Click the **Region** for **avaya**


System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Cisco Systems
Find and List Regions
Add a New Region
2 matching record(s) for Region Name begins with ""
Find Regions where Region Name begins with Find
and show 20 items per page
To list all items, click Find without entering any search text.
Matching record(s) 1 to 2 of 2
Region
avaya
Default
Delete Selected
First Previous Next Last
Page 1 of 1

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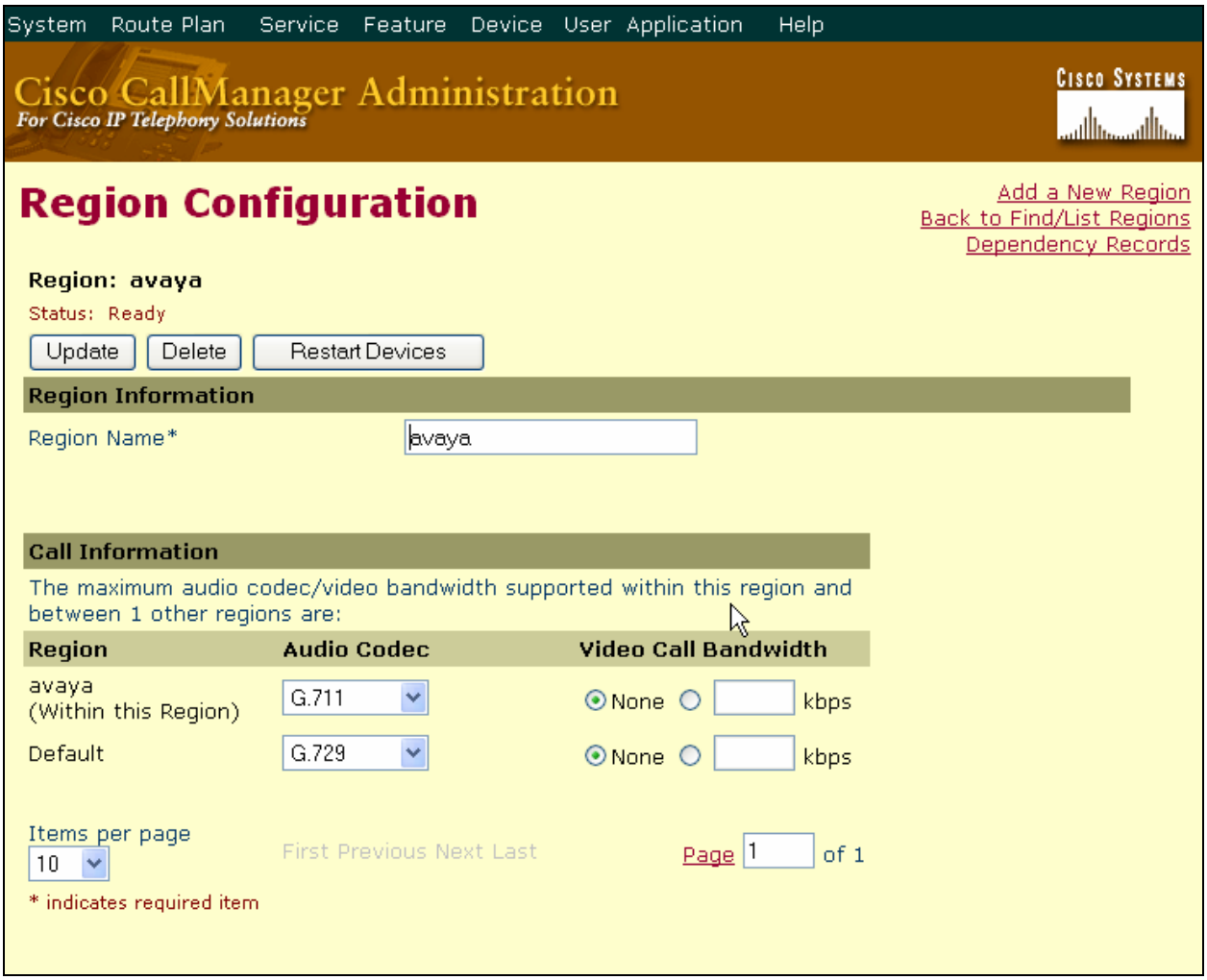
- Select **G.711** for region **avaya** and **G.729** for region **default** as shown below
- Click **Update**
- Click **Restart Devices**



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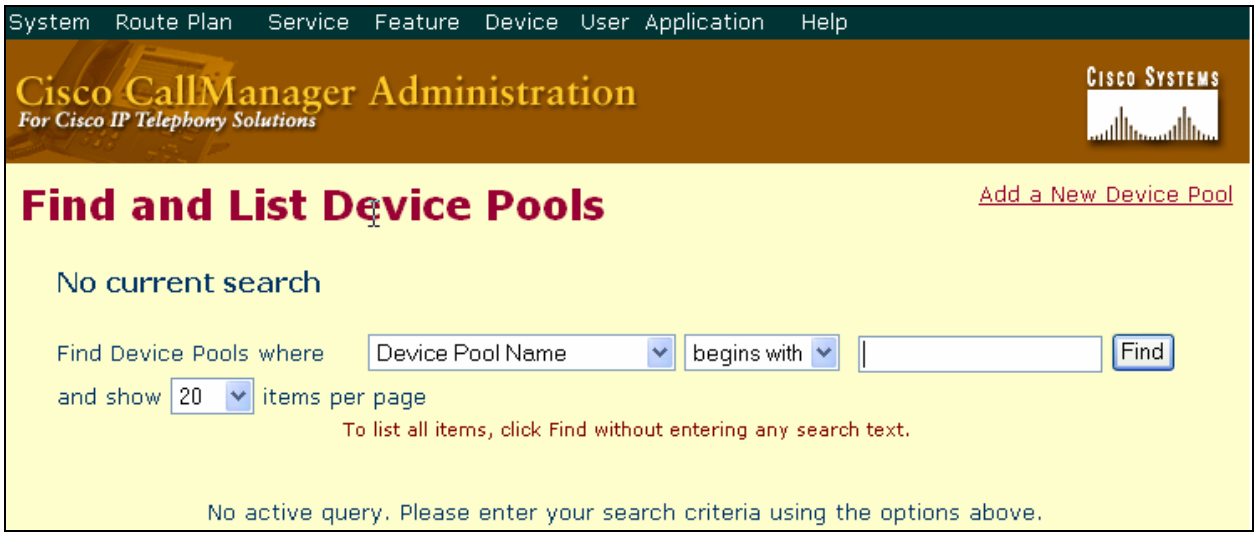
4.2.3. Add a New Device Pool

There is a default device pool pre-defined on the Cisco CallManager. In these Application Notes, the Cisco CallManager will use this default device pool. A new device pool, named avaya, will be created for Avaya S8300 Server with G350 Media Gateway in Location A. The purpose of creating a new device pool is to use different regions defined in previous section to select different codec. The display below shows the device pool configuration utilized for these Application Notes. Follow the following steps to add a new device pool to the Cisco CallManager database.

- Click **System → Device Pool**
- Click link **Add a New Device Pool**


System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Cisco Systems
Find and List Device Pools
Add a New Device Pool
No current search
Find Device Pools where
Device Pool Name
begins with
Find
and show
20
items per page
To list all items, click Find without entering any search text.
No active query. Please enter your search criteria using the options above.

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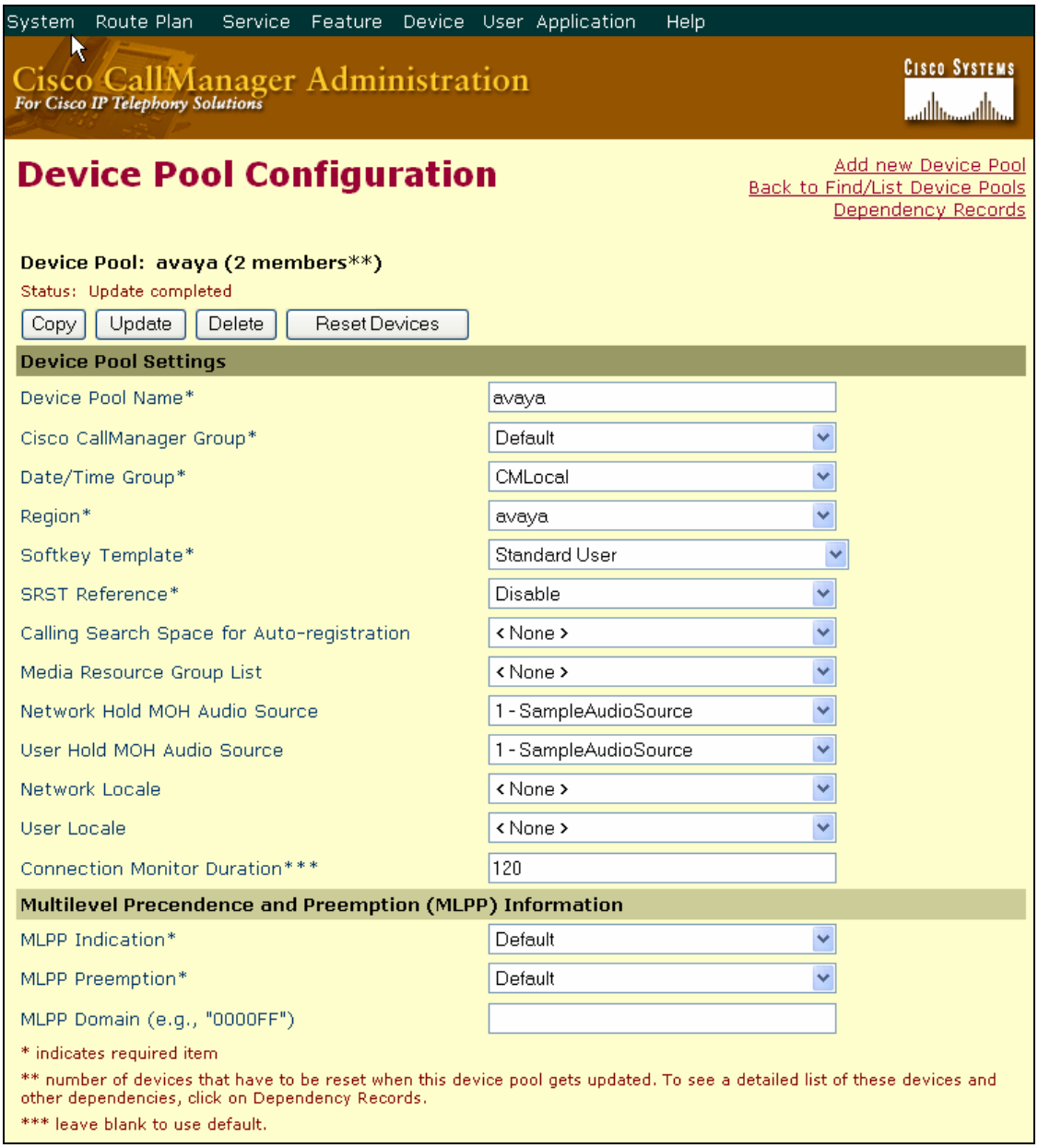
- Enter **avaya** as **Device Pool Name**.
- Use drop-down menu to select **avaya** as **Region**. This will put this device pool into region avaya, which will use G.729 codec for calls to other region.
- Use drop-down menu to select **1-SampleAudioSource** for **Network Hold MOH Audio Source**.
- Use drop-down menu to select **1-SampleAudioSource** for **User Hold MOH Audio Source**.
- Leave other fields as default.
- Click **Update**
- And Click **Reset Device**


System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Cisco Systems
Device Pool Configuration
Add new Device Pool
Back to Find/List Device Pools
Dependency Records
Device Pool: avaya (2 members**)
Status: Update completed
Copy Update Delete Reset Devices
Device Pool Settings
Device Pool Name* avaya
Cisco CallManager Group* Default
Date/Time Group* CMLocal
Region* avaya
Softkey Template* Standard User
SRST Reference* Disable
Calling Search Space for Auto-registration < None >
Media Resource Group List < None >
Network Hold MOH Audio Source 1 - SampleAudioSource
User Hold MOH Audio Source 1 - SampleAudioSource
Network Locale < None >
User Locale < None >
Connection Monitor Duration*** 120
Multilevel Precedence and Preemption (MLPP) Information
MLPP Indication* Default
MLPP Preemption* Default
MLPP Domain (e.g., "0000FF")
* indicates required item
** number of devices that have to be reset when this device pool gets updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.
*** leave blank to use default.

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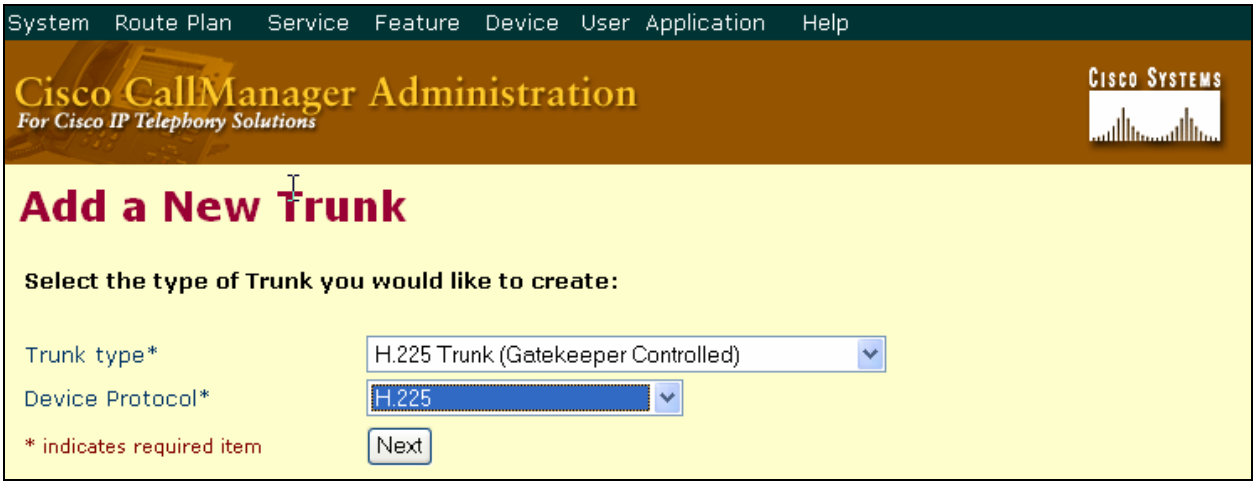
4.2.4. Add H.225 Trunk to the Gatekeeper

A H.225 trunk is required for the Cisco CallManager to communicate with the Cisco 2811 Gatekeeper. The displays below show the configuration of adding a H.225 trunk to Cisco 2811 Gatekeeper.

- Click **Device → Trunk → Add New Trunk**
- Using the drop-down menu to select **H.225 Trunk (Gatekeeper Controlled)**
- Select **H.255** as default for the **Device Protocol**
- Click **Next**



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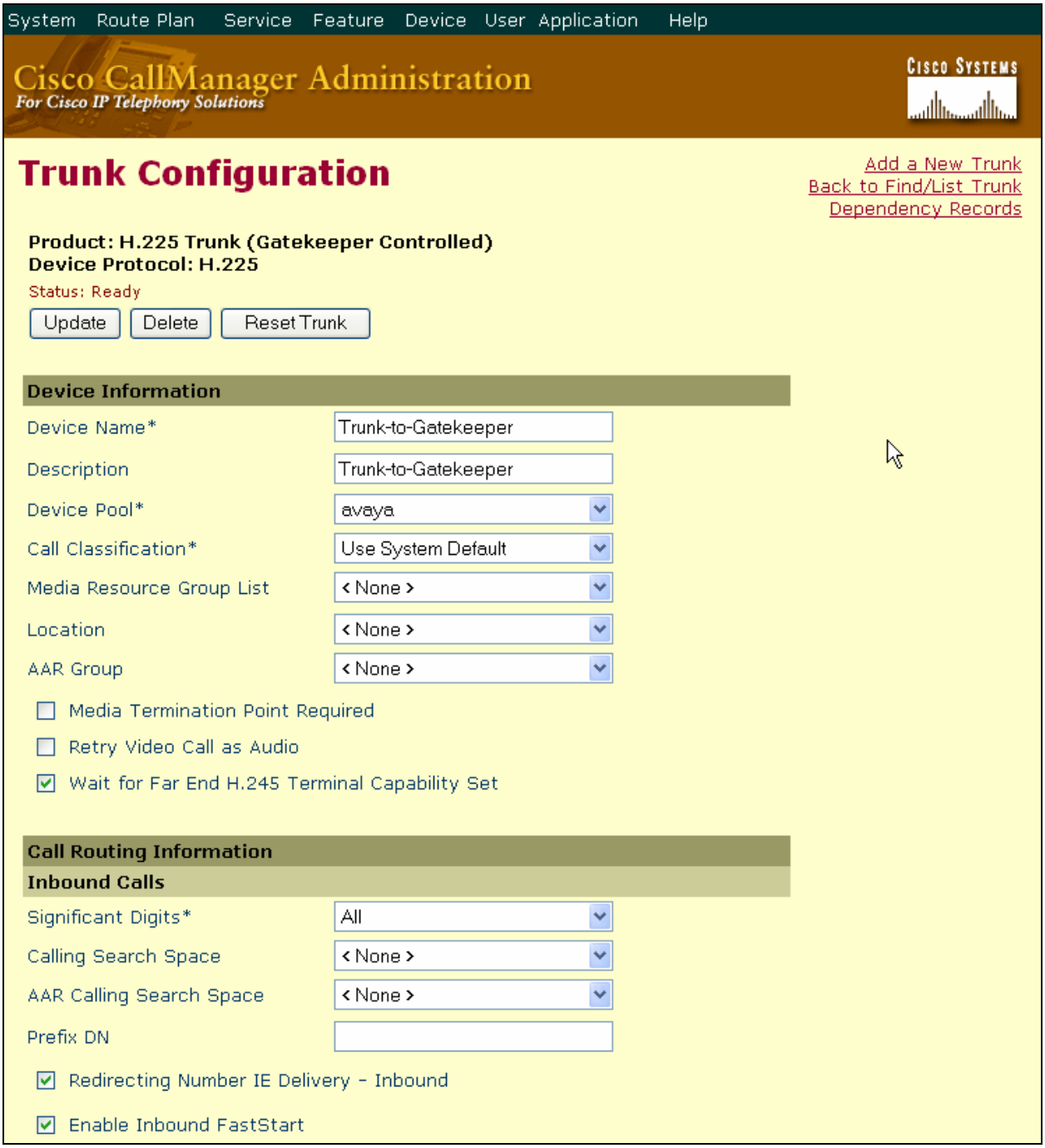
- Enter **Trunk-to-Gatekeeper** as **Device Name**
- Select **avaya** for **Device Pool**
- Check the box for **Wait for Far End H.245 Terminal Capacity Set**
- Check the box for **Enable Inbound FastStart**



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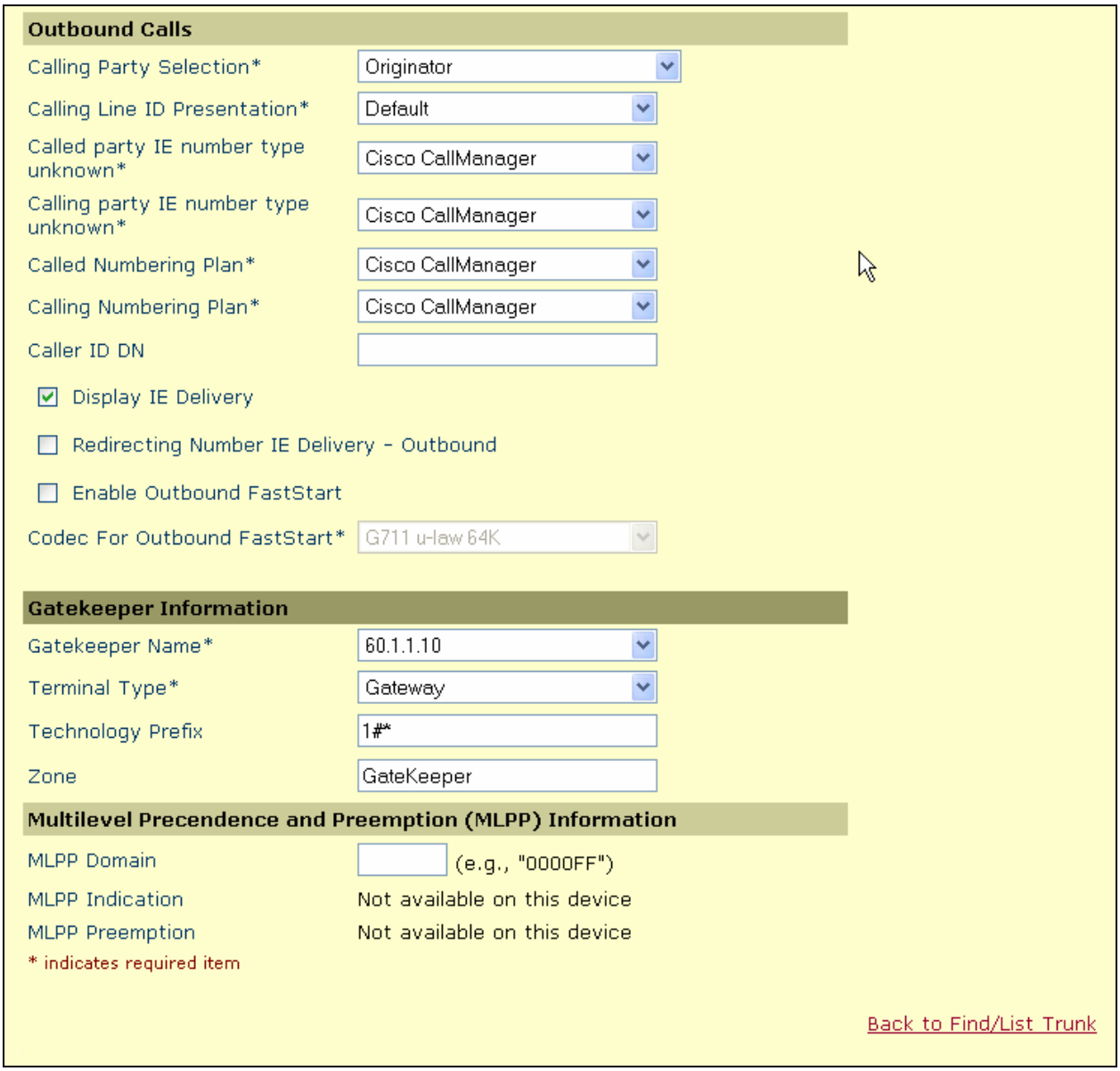
- Using the drop-down menu to select Gatekeeper IP address **60.1.1.10** for **Gatekeeper Name**
- Using the drop-down menu to select **Gateway** for **Terminal Type**
- Enter **1#*** as **Technology Prefix** to match the gateway prefix configured on Cisco 2811 Gatekeeper.
- Leave other fields as **default**
- Click **Update**
- Click **Reset Trunk**



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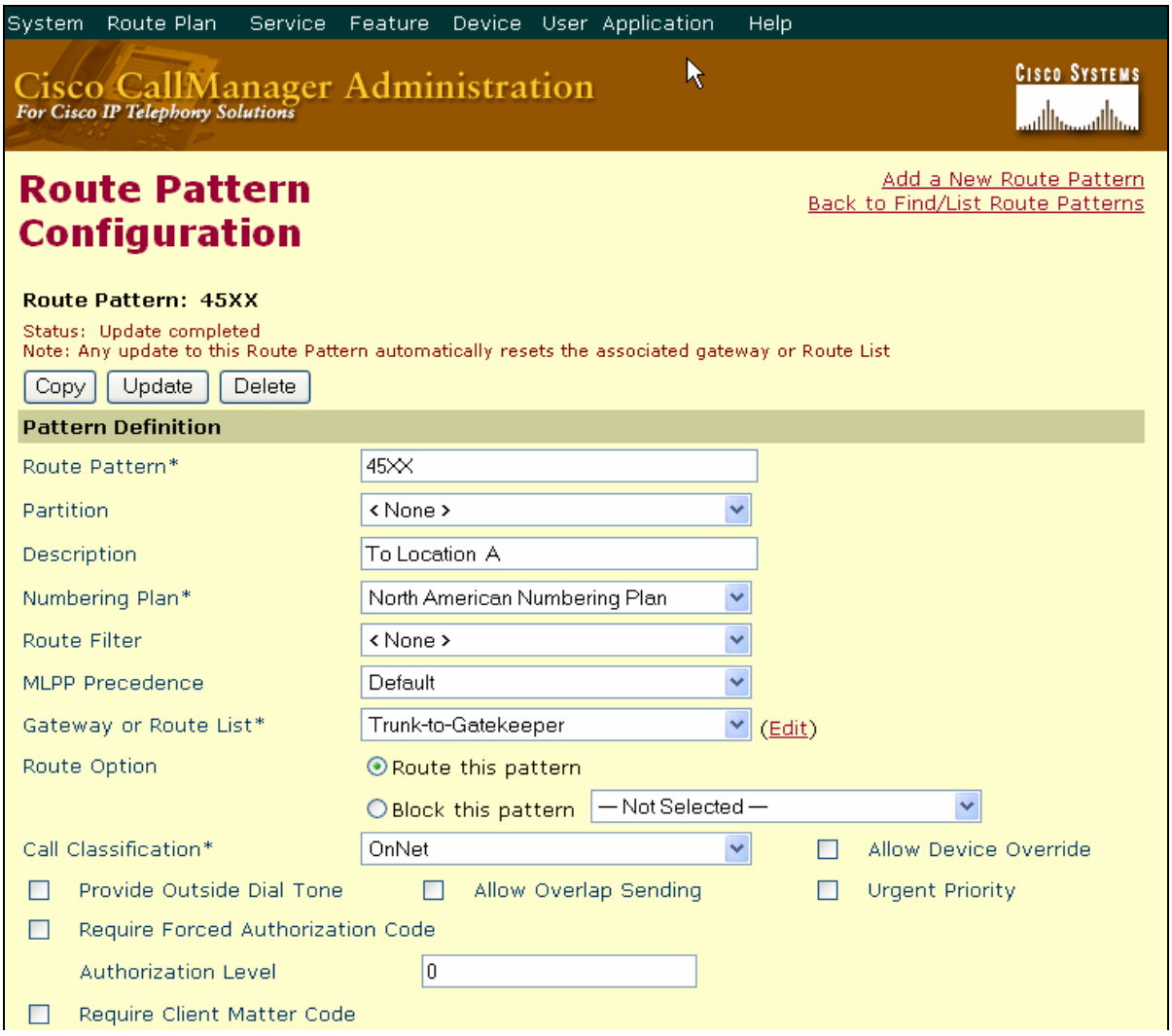
4.2.5. Configure a Route Pattern

Extension range 45xx is assigned to users at Location A. When users dial 45xx from the Location B, the Cisco CallManager knows how to route the calls to the Gatekeeper using the route pattern configured below. The call setup message will be carried out via the trunk selected in this route pattern. The display below shows the route pattern configuration utilized for these Application Notes. Follow the following steps to add a route pattern.

- Click **Route Plan → Route/Hunt → Route Pattern → Add a New Route Pattern**
- Enter **45XX** for **Route Pattern**
- Using drop-down menu to select **North American Numbering Plan** for **Numbering Plan**
- Using drop-down menu to select **Trunk-to-Gatekeeper** for **Gateway or Route List**
- Select the radio button for **Route this pattern**
- Leave remaining parameters as default setting
- Click **Update**



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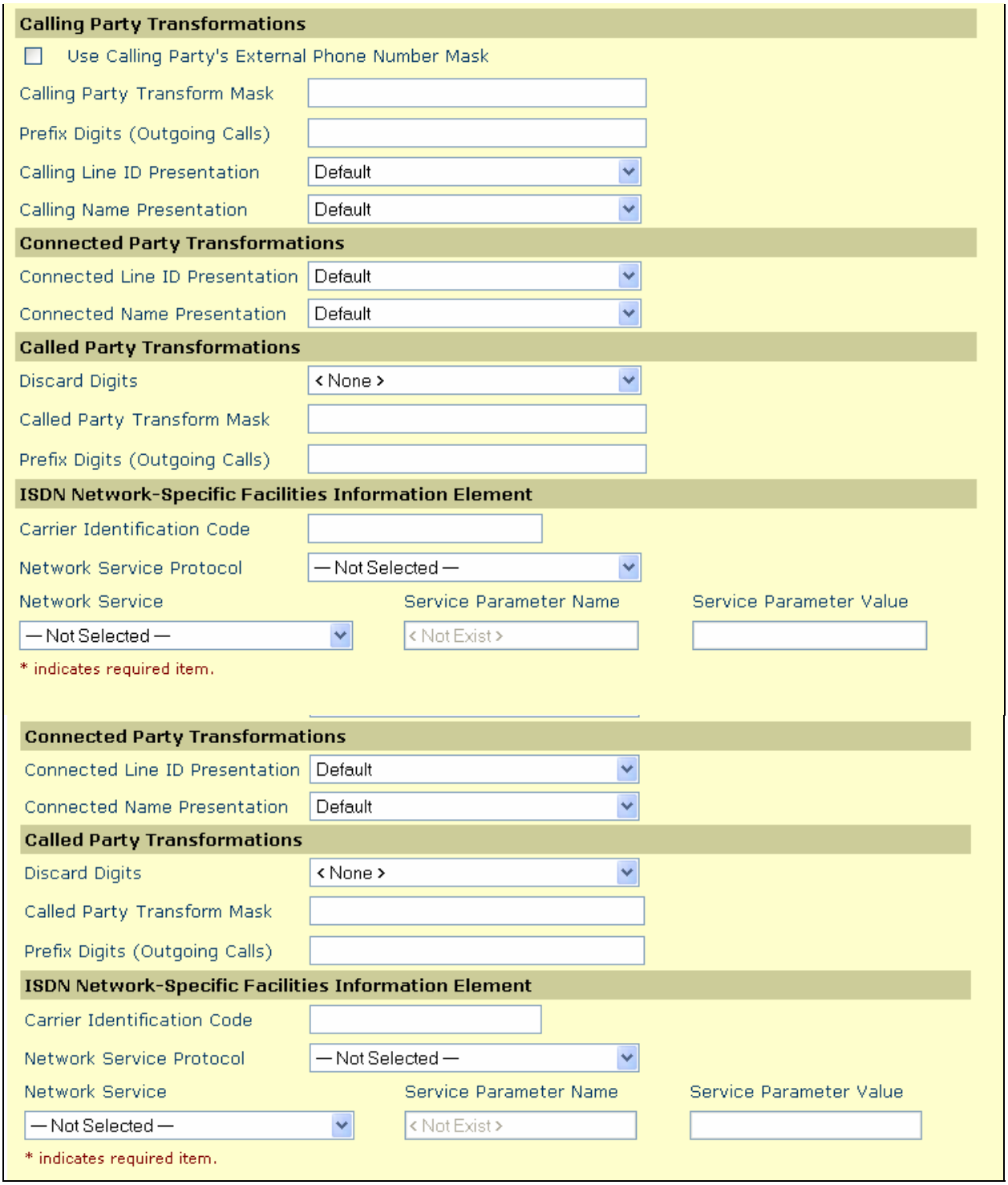

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5. Verification Steps

The following are the verification steps:

- Use the commands **status signaling-group 20** to check the signaling group is in service.

```
status signaling-group 20
                         STATUS SIGNALING GROUP

       Group ID: 20                               Active NCA-TSC Count: 0
     Group Type: h.323                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
    Group State: in-service
```

- Make a call from extension 61000 (Cisco CallManager phone) to extension 4555 (Avaya IP telephone). Use the command **show gatekeeper call** from the Cisco 2811 Gatekeeper and verify that this is a gatekeeper controlled call. Note that bandwidth (BW) is shown 16 (Kbps), which is the VoIP call with G.729 codec. Also, the call is between two endpoints extension 61000 and 4555 as shown below.

```
GateKeeper#show gatekeeper call

Total number of active calls = 1.
                         GATEKEEPER CALL INFO
                         ====================
LocalCallID                              Age(secs)   BW
542-195                                  6           16 (Kbps)
 Endpt(s): Alias                         E.164Addr
   src EP: Trunk-to-Gatekeeper_1         61000
           CallSignalAddr  Port  R ASSignalAddr    Port
           60.1.1.5        54155   60.1.1.5        59507
 Endpt(s): Alias                         E.164Addr
   dst EP:                               4555
           CallSignalAddr  Port    RASSignalAddr   Port
           22.1.1.22       1720    22.1.1.22       1719
```


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- Use the command **status station 4555** from the SAT interface on Avaya Communication Manager to check the status of this Avaya IP telephone. As shown from page 4 below, this is a call using codec **G.711MU** from extension 4555 to S8300 Media Gateway VoIP engine (**22.1.1.1**) and the connection-type is **ip-tdm.**

```
status station 4555                                          Page   4 of   7

                                AUDIO CHANNEL
                                Port: S00001
                  Switch                        IP                     IP
                  Port     Other-end IP Addr :Port   Set-end IP Addr:Port
G.711MU     Audio: 1         22.  1.  1.  1   :2050    22.  1.  1.222:61974
        Node Name:
   Network Region:          1                          1
   Audio Connection Type: ip-tdm

                          Port: S00001                Shared Port:
  Product ID and Release: IP_Phone   1.500
 H.245 Tunneled in Q.931? does not apply
     Registration Status: registered-authenticated
             MAC Address: 00:04:0d:ec:06:c6
      Native NAT Address: not applicable
ALG - NAT WAN IP address: not applicable
     Authentication Type: DES-56-plus
```

- Page 6 shows the same call from S8300 Server VoIP engine (**22.1.1.1**) to a Cisco phone (**60.1.1.100**) using codec g729.

```
status station 4555                                          Page   6 of   7
                     SRC PORT TO DEST PORT TALKPATH
src port: S00001
S00001:TX:22.1.1.222:61974/g711u/20ms
001V038:RX:22.1.1.1:2050/g711u/20ms   TX:ctxID:292
001V037:RX:ctxID:292   TX:22.1.1.1:2052/g729/20ms
T00091:RX:60.1.1.100:24204/g729/20ms
```

- Change bandwidth to 16 kbs on Gatekeeper for remote zone (Location A). Place two calls from Cisco CallManager to Location A and verify the fist call is successful and the second call was rejected by gatekeeper because no more bandwidth available.
- Verify that caller ID (calling name and number) can be displayed correctly.

6. Conclusion

As illustrated in these Application Notes, calls can be made between Avaya Communication Manager and Cisco CallManager via a Cisco 2811 Gatekeeper.

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